SIMULATION OF REVERBERATION FOR ACOUSTIC TEACHING

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Abstract

Reverberation is an important sound wave propagation process, which provoke the different human perceptions of the original sounds. It is very important for the students and other people learning acoustic to understand the reverberation phenomena in natural and artificial occasions.

Matlab is good instrument for doing reverberation simulation in all kinds of aspects: testing the different methods mathematically describing the reverberation, subjective perception and evaluation of each method etc.

The goal of this paper is to present the structure of programs for simulation of reverberation, that are build in Matlab. The steps, which each student can make, in order to follow the simulation algorithm and to understand the characteristics and applications of artificial reverberation are shown. Some graphical and visualization means to improve the sound wave human perception are also presented.

1. Introduction

The phenomena "reverberation" exists in the concert halls recording rooms or in the other places, where the music or other sounds are played [1]. It is very important to know the sound wave propagation in such halls. If there are no walls or other objects, the sound waves consist only off the direct wave from the source. But this is an unreal situation. There are always reflected sound waves, which are added with direct waves. Therefore the resulting sound wave in each point of the hall is a sum from direct sound wave and an infinite number of reflected sound waves from walls, objects etc.

In a general case it is impossible to know, to calculate or to predict the amplitude and phase of the resulting sound wave in an arbitrary point in a hall. But there are some models and algorithms, which give the possibilities to present the sound wave image in a hall with some admissible tolerances and errors, respectively [2]. To teach students of these methods and algorithms is very usefully and necessary. The students learn the mathematical representation of each method for reverberation modeling. The work with a real programming system gives them the ability for practical simulation of different hall reverberation situation. The students can also better understand the reverberation methods, estimating the sound quality.

The other reason to teach the student of algorithms for reverberations is that the knowledge of these methods let the students realize many special effects using an artificial reverberation system in a concert hall. This is very important for the sound producers and sound operators.

2. Matlab simulation of reverberation algorithms

2.1. Reverberation algorithm

Figure 1 presents a possible reverberation algorithm [3].

The input signal is first put to a Predelay block. After that it is necessary to have a Multitap delay line with a set of n-outputs Tap 1L, Tap 1R, Tap 2L, Tap 2R, …, Tap nL, Tap nR for left and right stereo sound channels, respectively.

The important parts of each the reverberation algorithm are the blocks of all-pas and comb filters (EQ filters) for recirculation of the early reflections.

The reverberation output (Reverb output) is a mixed sound signal from the early reflections (amplified) and late decay.

It is shown in Fig.1, that in the algorithms, for an extension of the decay time it is necessary to have a Feedback path (Fig. 2).
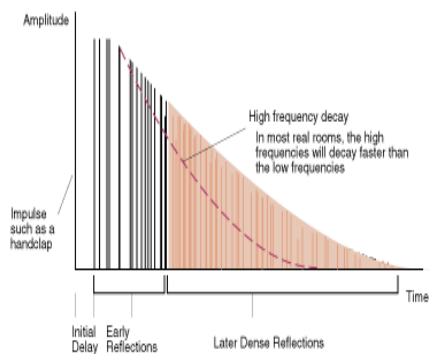

Figure 2. Decay time

2.2. Filters representation

In the reverberation algorithm, which is present in Figure.1 the blocks of the filters are present. Some of them are low pass, other – all pas filters. All of them use the output of the Multitap delay line. Its transfer function can be written as:

$$H(z) = z^{-k} \qquad (1)$$

The low pass filter transfer function is given by the expression:

$$H(z) = \frac{1 - g}{1 - gz^{-1}} \qquad \Box (2)$$

The low pass filters are used to simulate the air absorption. The gain of the filter is:

$$g = 2 - \cos\left(2\pi \frac{f_c}{f_s}\right) - \sqrt{\left(\cos\left(2\pi \frac{f_c}{f_s}\right) - 1\right)^2 - 1} \qquad (3)$$

The low pass filter block diagram used in the Matlab Simulation of the reverberation algorithm is shown in Figure 3.
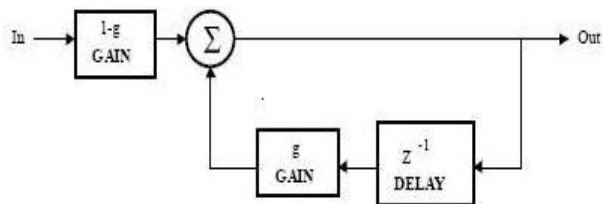

Figure 3. Low-pass filter

2.3. Matlab simulation

All of the blocks presented in algorithm of Figure 1 are simulated with Matlab.

There are two ways to make this simulation: as a Matlab program or as a Simulink block diagram representation. Each of these methods has some advantages.

The simulation of the reverberation algorithm as a program allows the students to understand each step of the algorithm precisely, making the program by themselves.

Simulink representation of reverberation algorithm is useful in other direction – there it is possible to make quick changes and to analyze the results with the help of the embedded visualization.

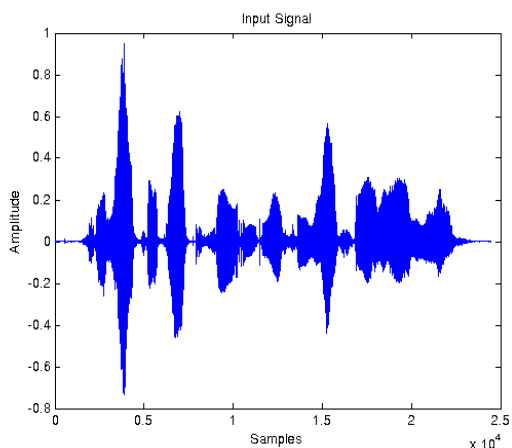The results of the simulations are given in the Figure 4 and Figure 5.
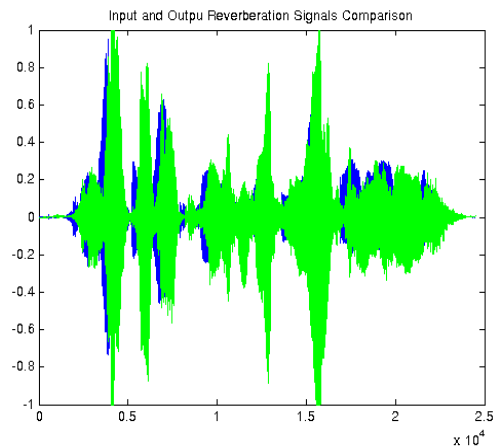

Figure 4. Input Signal

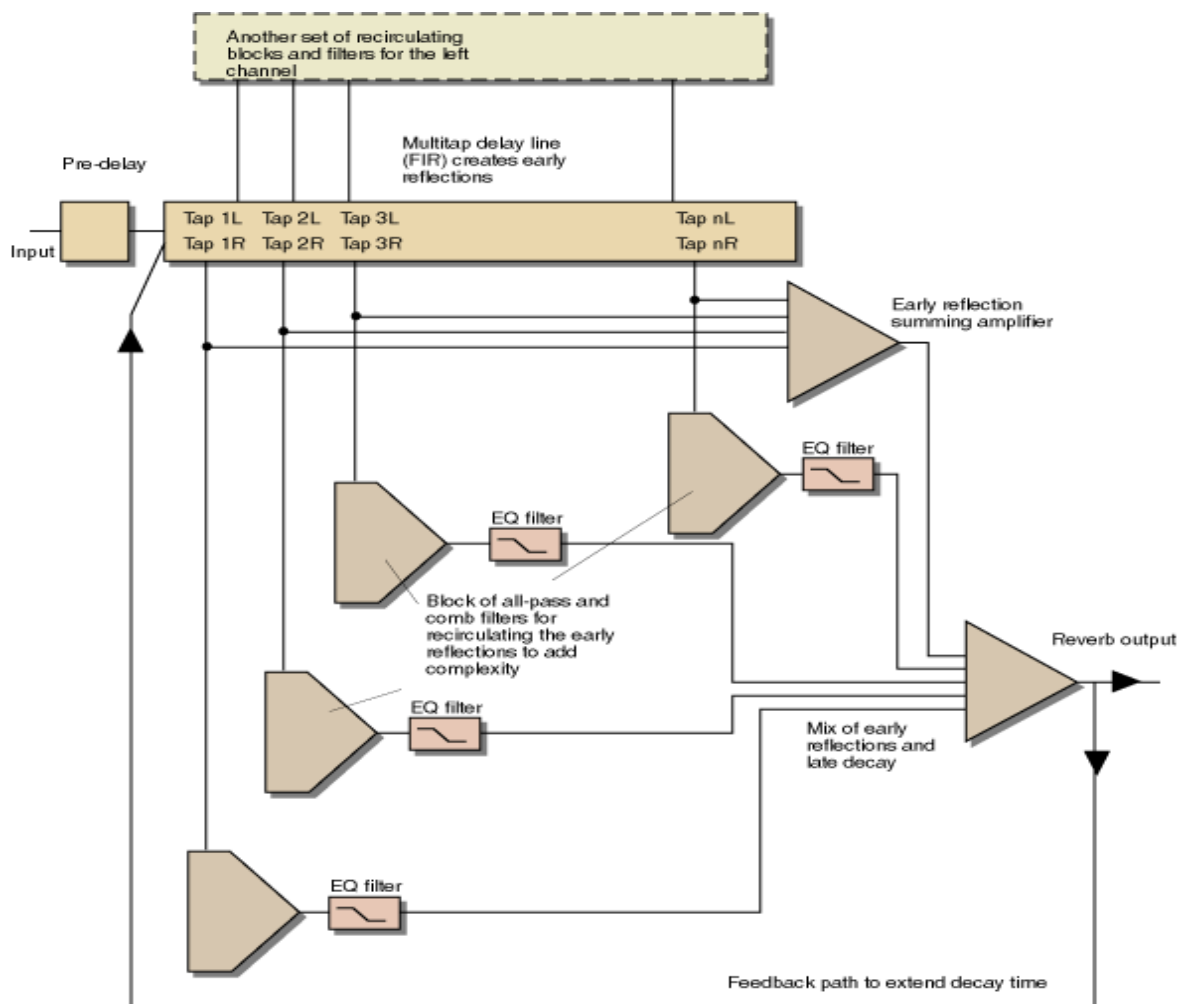**Figure 5.** Input and Output Signal

Figure 1. Reverberation algorithm

3. Conclusion

In the conclusion, it could be summarized that the proposed Matlab simulation of reverberation algorithms in this paper provides a good tools for practical education of the students learning acoustics, for the reverberation is an important part of this course.

References

[1]. M. R. Schroeder, "Natural Sounding: Artificial Reverberation", J. Audio Engineering Society. Vol. 10, No. 3, 1961.

[2]. W. G. Gardner, "Applications of Digital Signal Processing to Audio and Acoustic. Chapter 3. Reverberation Algorithms", Kluwer Academic Publishers, 1998.

[3]. J. Dattorro, "Effect Design. Part 1: Reverberation and Other Filters", J. Audio Engineering Society. Vol.45, No. 9, 1997.